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(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number : **0 501 690 A2**

(12) **EUROPEAN PATENT APPLICATION**

(21) Application number : **92301462.5**

(22) Date of filing : **21.02.92**

(51) Int. Cl.^{5} : **H04J 1/00**, H04B 7/185

(30) Priority : **28.02.91 GB 9104186**

(43) Date of publication of application : **02.09.92 Bulletin 92/36**

(84) Designated Contracting States : **AT BE DE DK ES FR GB IT NL SE**

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(54) **Apparatus for and method of digital signal processing.**

(57) Apparatus for and a method of digital signal processing utilises a technique known as sub-band decomposition and reconstruction. A plurality of input channels are provided each of which is mixed with a vernier frequency correction. In a multiplexer implementation, an FFT weighted overlap-add demultiplexer (c) is provided for decomposing each input channel into sub-bands. The sub-band outputs are decimated by a factor (d) to the common input sampling rate of a multiplexer, the sub-bands from each channel then being passed into the appropriate ports of an FFT weighted overlap-add multiplexer (f) by means of a switch matrix (e). The multiplexer (f) reconstructs the individual channels, interpolates each channel up to the output sampling rate of the FDT signal, mixes each channel up to its assigned carrier frequency in the FDM and multiplexes the channels. This provides a computationally efficient architecture with the flexibility to accommodate channels of differing bandwidths.

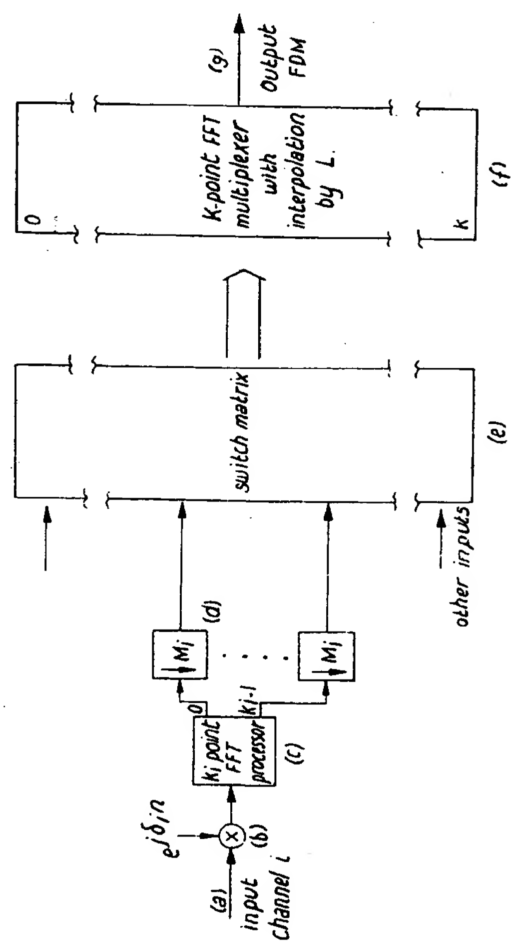

Fig. 1

EP 0 501 690 A2

The present invention relates to apparatus for and a method of digital signal processing.

In the specification the following abbreviations are employed:-

ASIC: Application Specific Integrated Circuit
DSP: Digital Signal Processing
FDM: Frequency Division Multiplex
FIR: Finite Impulse Response
FFT: Fast Fourier Transform
MCDD Multi-Carrier Demultiplexer/Demodulator
OBP: On-Board Processing

This invention is in the field of DSP ASIC architectures for frequency multiplexing and demultiplexing of sampled signals. A considerable body of work has already been done on such circuits which form the central component of OBP payloads proposed for a wide variety of near-term satellite communications systems. The aim is to demultiplex or multiplex an FDM of signal channels on-board the satellite for purposes which can include: individual channel power control and/or channel to beam routing and/or subsequent demodulation of the signal channels in an MCDD.

The focus is always on reducing the computational complexity of the architecture, and hence the ASIC mass and power requirements. Many efficient architectures are based on the use of an FFT to simultaneously demultiplex or multiplex a block of signal channels. A good example of the current "state-of-the-art" in such designs is described in the specification of United Kingdom Patent Application No. 9005178 dated 8th March, 1990 in the name of the Applicants.

Existing designs based on block-FFT processing impose the constraint of a uniform channel stacking scheme, that is, the individual channel slots must be equally spaced and contiguous see R.E. Crochiere and L.R. Rabiner, 'Multi-Rate Digital Signal Processing', Prentice-Hall, 1983 .This is because the FFT acts as a uniform filter bank. This constraint can be a disadvantage; many attractive system scenarios require multiplexing/ demultiplexing of channels which have a mix of different bandwidths. For example, it may be desirable to alter the bandwidth spacings of the channels processed in the OBP after the satellite is in operation, in response to a change in traffic demand.

An architecture which aims to offer this flexibility is described in S.J. Campanella and S. Sayegh, 'A Flexible On-Board Demultiplexer/Demodulator', Comsat laboratories, which is based on the well known overlap-save technique for digital FIR filtering using an FFT (see also R.E. Crochiere and L.R. Rabiner, 'Multi-Rate Digital Signal Processing' Prentice-Hall, 1983). This design, however, appears to ignore an essential facet of this technique, namely that the length of the overlap has to be precisely one sample less than the length of the unit sample response of the FIR filter (see R.E. Crochiere and L.R. Rabiner, 'Multi-Rate Digital Signal Processing' Prentice-Hall, 1983). The practical effect of this oversight would be to degrade the performance of the proposed system (in terms of signal quality) by adding a noise-like distortion to the demultiplexed channel. The design could be modified to be mathematically correct but this would seriously compromise its computational efficiency.

An object of the present invention is to mitigate this problem.

According to one aspect of the present invention there is provided apparatus for digital signal processing for variable bandwidth signals comprising a bank of contiguous digital filters having overlapping frequency responses which together span the bandwidth of the input signal and sum to unit allpass response.

According to another aspect of the present invention there is provided a method of digital signal processing including the steps of feeding an input signal to a bank of contiguous digital filters having overlapping frequency responses which together span the bandwidth of the input signal and subsequently summing individual filter outputs to reconstruct the input signal.

In order that the invention may be more clearly understood, one embodiment of the invention will now be described, by way of example, with reference to the accompanying drawings, in which:-

Figure 1 shows a block FFT multiplexer for variable bandwidth inputs, and

Figures 2a to 2d show the frequency response of filters for the multiplexer of Figure 1.

The invention utilises a technique known as sub-band decomposition and reconstruction. This technique is described in A. Papoulis, 'Signal Analysis', McGraw-Hill, 1984; M.R. Portnoff, 'Implementation of the digital phase vocoder using the fast Fourier transform' IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-24, pp243-248, June 1976; R.E. Crochiere, 'A Weighted Overlap-Add Method of Short-Time Fourier Analysis/Synthesis', IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-28, pp 99-102, February, 1980. In effect this is a computationally efficient implementation of a bank of contiguous digital filters which have overlapping frequency responses which together span the bandwidth of the input signal and sum to unit response. This means that the individual filter outputs, or sub-bands, can be subsequently summed to reconstruct the input signal. If the sub-band signals are decimated (downsampled) then it is necessary to interpolate

them prior to reconstruction, which functions are efficiently combined in an FFT multiplexer (see M.R. Portnoff, 'Implementation of the digital phase vocoder using the fast Fourier transform' IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-24, pp243-248, June 1976, R.E. Crochiere, 'A Weighted Overlap-Add Method of Short-Time Fourier Analysis/Synthesis', IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-28, pp 99-102, February, 1980).

The multiplexer aspect of the present invention is shown in Figure 1. The processing stages are as shown, namely:

1. each-baseband video signal input channel (a) is mixed with a vernier frequency correction (b)
2. each input channel is decomposed into sub-bands using an FFT weighted overlap-add demultiplexer (c), with a suitable filter design
3. for each channel, the sub-band outputs are decimated (d) to the common input sampling rate of the multiplexer
4. the sub-bands from each channel are now passed into the appropriate ports of an FFT weighted overlap-add multiplexer (f) by means of a switch matrix (e); the processor (f) performs the function of
   - reconstructing the individual channels
   - interpolating each channel up to the output sampling rate B of the FDM signal
   - mixing each channel up to its assigned carrier frequency in the FDM
   - multiplexing the channels

The block-FFT multiplexer (f) features a K-point complex-complex FFT transform and interpolation by a factor L. The multiplexer output (g) is a complex baseband video signal comprising the FDM. The individual input channels (a) are bandlimited complex baseband video signals. An input channel i is divided into K_i sub-bands using a K_i-point FFT demultiplexer (c); each sub-band output from the channel i demultiplexer is decimated by a factor M_i (d). Frequencies w (in radians/sample) in the input channel are therefore mapped to multiplexer output frequencies w′ by the relation:

$$w \rightarrow w' = w M_i/L \quad (1)$$

The sub-bands corresponding to channel i are spaced by $\Delta w = 2\pi/K_i$. Proper reconstruction of a channel i in the output multiplex requires that:

$$\Delta w \rightarrow \Delta w' = > (2\pi/K_i)(M_i/L) = 2\pi/K => K/K_i = L/M_i \geqq 1 \quad (2)$$

Figure 2a shows the prototype frequency response of the filter $H_i(e^{jw})$ associated with sub-band decomposition of the channel i using a K_i-point FFT. The ideal response (I) is a 'brick-wall' filter; the practical response (P) is relaxed, with stopband w_s less than $2\pi/K_i$ as shown. Note that this implies that the decimation faction M_i associated with the sub-bands of channel i must be less than K_i.

Figure 2b shows the prototype frequency response of the common filter $F(e^{jw'})$ associated with the subsequent reconstruction of the channel sub-bands and interpolation up to the FDM output frequency in the multiplexer; the response shown is for the particular case of interpolation factor L = K/4 (practical filter designs require that L < K). The ideal anti-image (interpolation) filter (I) associated with the FFT multiplexer would be 'brick-wall' filter with passband edge at $w' = \pi/L = 4\pi/K$. In practice this must be relaxed as shown, with the passband edge of the interpolating filter (P) at $w'_p = 2\pi/K$ greater than or equal to the stopband edge of the sub-band decomposition filter. The frequency response of the filter $H_i(e^{jw})$ is shown in Figure 2b as it maps onto the output frequency axis w′ according to the relations (1) and (2) above.

Note that:

$$|F(e^{jw'})| = 1 \text{ for } 0 \leqq |w'| \leqq 2\pi/K \quad (3)$$

This, combined with the use of a standard windowed FIR design for the sub-band decomposition filters H_i, which preserves the zero-crossings of a rectangular window in the unit sample response,satisfies the requirements for correct reconstruction, see M.R. Portnoff, 'Implementation of the digital phase vocoder using the fast Fourier transform' IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-24, pp243-248, June 1976.

The input channel i is mixed to a centre frequency:

$$w'_i = \delta_i + 2\pi k/K \quad (4)$$

where k is an integer O,...,K-1 and δ_i a vernier frequency offset supplied by the channel mixer (b) in Figure 1, where:

$$|\delta_i| \leqq \pi/K \quad (5)$$

In the example being discussed the interpolation factor in the multiplexer is L = K/4 and so the sub-bands input to the FFT multiplexer must be at a sampling rate of $2\pi/L = 8\pi/K$.

It can be seen from Figure 2c that an input channel i which is bandlimited to less than $\Delta w' = 2\pi/K$ (and hence is oversampled by at least a factor of 4) does not require sub-band decomposition. The spectrum of channel i is denoted as $X(e^{jw'})$. Even with the maximum vernier shift of $|\delta_i| = \pi/K$ the shifted spectrum $X(e^{j(w'-\pi/K)})$ of channel i still lies entirely within the passband of the anti-image filter F. Such channels can therefore be directly entered into the switch matrix bypassing processing blocks (c) and (d) of Figure 1.

Figure 2d shows the case where an input channel i has a bandwidth > $\Delta w'$ and is subjected to a vernier shift of π/K. The channel must be decomposed into sub-bands to avoid distortion in the transition band of the anti-image filter F and leakage from the images at $w' = -2\pi/L$.

This system can be designed to accommodate a particular mix of input channel bandwidths; with appropriate mixing and switching of the sub-band groups to the input ports of the multiplexer these input channels can be multiplexed to different bands of the output FDM. With programmable FFT processors (c) on each input line this system offers the flexibility to be reconfigured at a later date to accommodate a different mix of bandwidths in the input channels.

The demultiplexer is simply the multiplexer run in reverse.

The above described embodiment provides a mathematically exact and computationally efficient architecture with the flexibility to accommodate channels of differing bandwidths. In particular the performance of the multiplexer/demultiplexer in terms of its impact on signal quality can be quantified exactly given the design specifications on the filters used.

It will be appreciated that the above embodiment has been described by way of example only and that many variations are possible without departing from the scope of the invention.

Claims

1. Apparatus for digital signal processing for variable bandwidth signals comprising a bank of contiguous digital filters having overlapping frequency responses which together span the bandwidth of the input signal and sum to unit response.

2. Apparatus for digital signal processing as claimed in claim 1, comprising a plurality of signal input channels (a) each of which is mixed with a vernier frequency correction (b).

3. Apparatus for digital signal processing as claimed in claim 1 or 2, comprising a plurality of signal input channels (a) each of which is decomposed into sub-bands using a FFT weighted overlap-add demultiplexer(b).

4. Apparatus for digital signal processing as claimed in claim 3, in which, for each channel, means (d) are provided for decimating the sub-band outputs by a factor to the common input sampling rate.

5. Apparatus for digital signal processing as claimed in claim 4, comprising an FFT weighted overlap-add multiplexer (f) and a switch matrix (e) operative to pass the sub-bands from each channel into appropriate ports of the multiplexer.

6. Apparatus for digital signal processing as claimed in claim 5, in which the FFT weighted overlap-add multiplexer (f) comprises a transform and interpolation by a factor.

7. Apparatus for digital signal processing as claimed in any of claims 2 to 6, in which the individual input channels are in complex baseband video form.

8. A method of digital signal processing including the steps of feeding an input signal to a bank of contiguous digital filters having overlapping frequency responses which together span the bandwidth of the input signal and subsequently summing individual filter outputs to reconstruct the input signal.

9. A method of digital signal processing as claimed in claim 8, in which, when the filter outputs are decimated, they are interpolated prior to reconstruction.

10. A method of digital signal processing as claimed in claim 8 or 9, in which there are a plurality of input channels (a) and each input channel is mixed with a vernier frequency correction (b).

11. A method of digital signal processing as claimed in claim 8, in which there are a plurality of input channels (a) and each input channel is decomposed into sub-bands using an FFT weighted overlap-add demultiplexer (f).

12. A method of digital signal processing as claimed in claim 11, in which, for each channel, the sub-band outputs are decimated by a factor to the common input sampling rate of a multiplexer.

13. A method of digital signal processing as claimed in claim 12, in which the sub-bands from each channel are passed into the appropriate ports of an FFT weighted overlap-add multiplexer (f) by means of a switch matrix (e), the multiplexer performing the function of reconstructing the individual channels, interpolating each channel up to the output sampling rate of the FDM signal, mixing each channel up to its assigned carrier frequency in the FDM and multiplexing the channels.

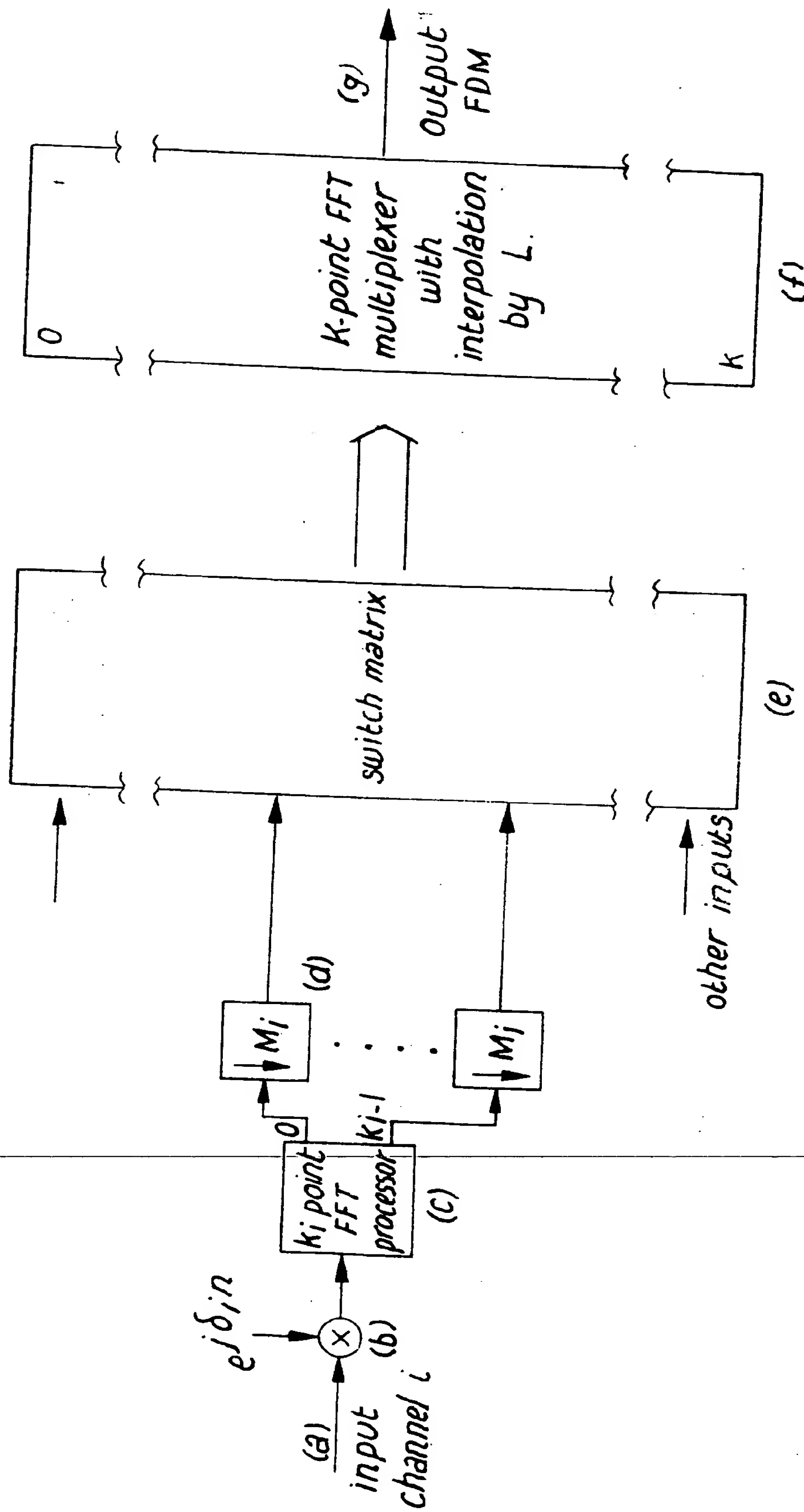
(a)
input
channel i
$e^{j\delta_i n}$
(b)
k_i point
FFT
processor
(c)
0
k_i-1
$\downarrow M_i$
(d)
$\downarrow M_j$
switch matrix
(e)
other inputs
0
k
K-point FFT
multiplexer
with
interpolation
by L.
(f)
(g)
Output
FDM

Fig. 1

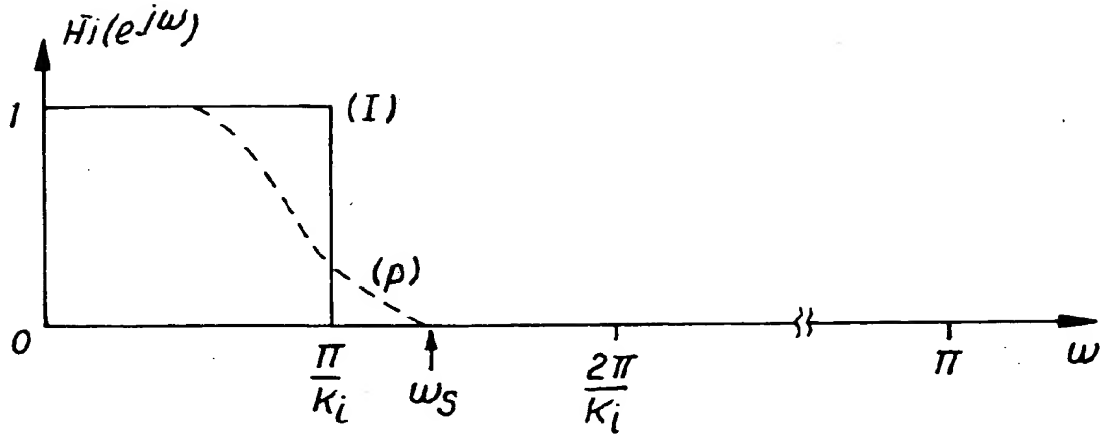
$H_i(e^{j\omega})$
1
(I)
(p)
0
$\frac{\pi}{K_i}$
ω_S
$\frac{2\pi}{K_i}$
π
ω

FIG. 2a

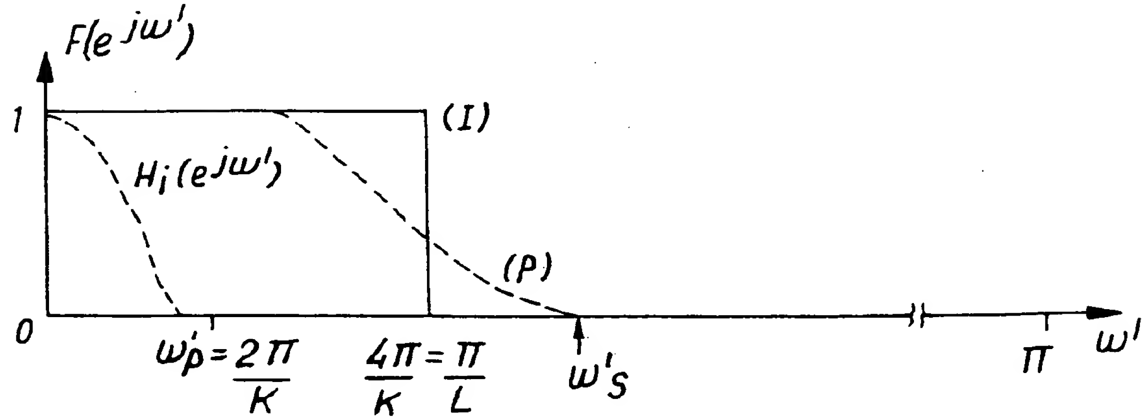
$F(e^{j\omega'})$
1
(I)
$H_i(e^{j\omega'})$
(p)
0
$\omega'_p = \frac{2\pi}{K}$
$\frac{4\pi}{K} = \frac{\pi}{L}$
ω'_S
π
ω'

FIG. 2b

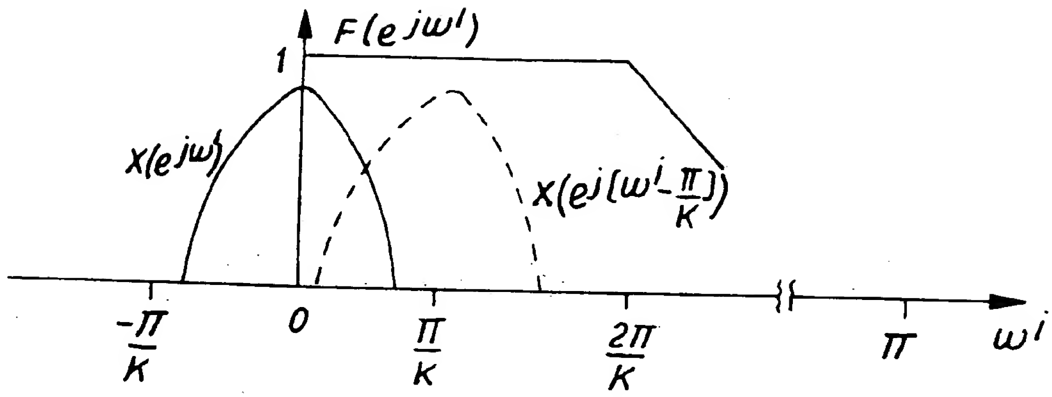
F(e jω')
1
X(e jω')
X(e j[ω' - π/K])
-π/K
0
π/K
2π/K
π
ω'

FIG. 2c

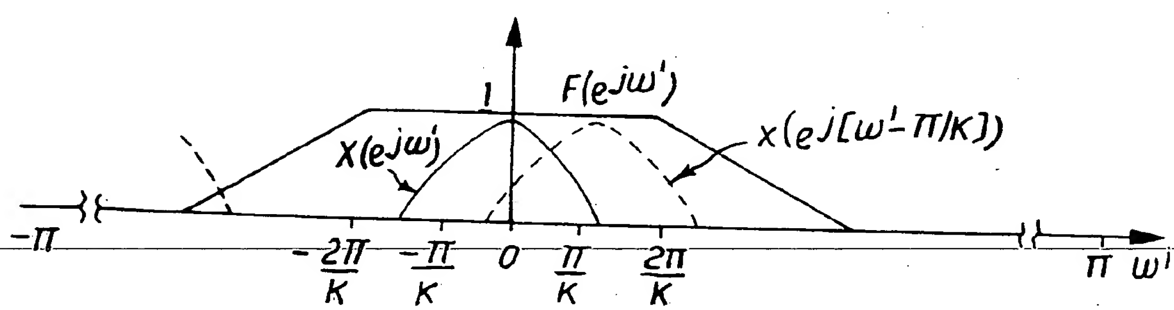
1
F(e jω')
X(e jω')
X(e j[ω' - π/K])
-π
-2π/K
-π/K
0
π/K
2π/K
π
ω'

FIG. 2d